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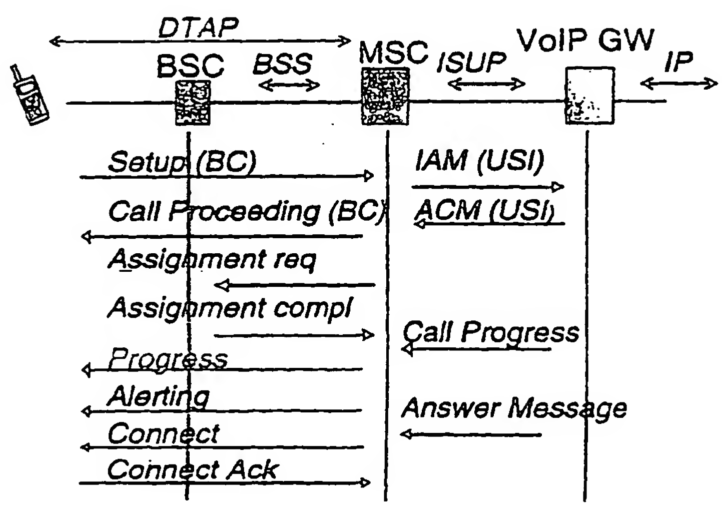

Signalling sequences for TFO over IP

(57) Abstract

The present invention relates to an arrangement for improving the speech quality, especially for VoIP (Voice over IP) calls, which arrangement comprises a Transceiver and Rate Adapter Unit (TRAU) in which an encoded speech signal from a Mobile Station (MS) is transcoded, and for the purpose of reducing the necessary encodin/decoding for thereby avoiding deterioration of speech quality, and also for the purpose of avoiding reduction in bandwidth, it is according to the present invention suggested that said arrangement comprises means for either putting the TRAU in a transparent mode or letting the TRAU be bypassed altogether.

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ARRANGEMENT FOR IMPROVING THE SPEECH QUALITY, ESPECIALLY FOR VoIP (VOICE OVER IP) CALLS

Field of the invention

The present invention relates to an arrangement for improving the speech quality, especially for VoIP (Voice over IP) calls, which arrangement comprises a Transceiver and Rate Adapter Unit (TRAU) in which an encoded speech signal from a Mobile Station (MS) is transcoded.

General background of the invention

This is a proposal for how the speech quality can be enhanced for mobile VoIP calls. It is a well known problem that speech can be degraded for mobile calls when too many speech encodings/decodings are performed on the voice path.

There are mobile standards to overcome this for MS to MS calls, called Tandem Free Operation (TFO). The Japanese Personal Digital Cellular (PDC) system uses outband Mobile Application Part (MAP) signalling, called codec through. In Global System For Mobile Communication (GSM) there is an emerging standard, TS 04.53, while there is no TFO planned for Digital-Advanced Mobile Phone Service (D-AMPS).

These TFO standards were developed for MS to MS calls, and they do not address in an optimal way TFO over IP.

The solution described here has the following main benefits:

- Improved speech quality for mobile VoIP calls compared to ordinary mobile calls

- The impacts are local affecting only the MS - IP side of a MS - IP - MS call

- The implementation uses outband standard #7 signalling (a standard in the CCITT Signalling system) and has minor impacts on the Mobile Switching Centre (MSC) and the Base Station Subsystem (BSS)

- It can be used with all existing mobile voice codecs as long as they are supported by the IP network to which the gateway is connected

- Avoids the use of transceivers in Base Station Subsystem

- No need for speech coding and the use of Digital Signalling Processors (DSP) in the VoIP gateway

Tandem Free Operation (TFO) in GSM

In case of MS to MS calls in a mobile network without TFO, the speech signal is encoded within the first mobile station for transmission on the air interface, and transcoded within the first Tranceiver and Rate Adapter Unit (TRAU). The Pulse Code Modulation (PCM) samples are then transported within the fixed part of the network to the second TRAU using 64 kbit/s traffic links. The second TRAU encodes the speech signal a second time for the transmission on the second air interface. The two codecs of the connection are in "Tandem Operation".

This Tandem Operation has several disadvantages:

- The extra encoding/decoding degrades the speech quality more than necessary

- The links between the TRAUs need 64 kb/s while 16 or 8 kb/s would be sufficient

- The unnecessary encoding/decoding within the TRAUs allocates Digital Signalling Processor (DSP) power

The European Telecommunications Standards Institute (ETSI) is working on a standard for TFO, TS 04.53. This standard defines inband signals between TRAUs so that TFO effects only the TRAUs and are therefore fully compatible with existing equipment.

In Fig. 1 there is illustrated in a schematical manner, the principle of Tandem Free Operation (TFO), and in the following there will be give a brief description of this principle.

TFO in GSM is defined as an inband signalling protocol between two peer transceivers. It

- Tests the path for possible TFO

- Establishes the TFO connection

- Guarantees a fast fallback procedure

- Supports resolution of Codec mismatch

The standard defines both TFO Frames (speech) and TFO messages. TFO Frames affects only the TRAUs.

For Half Rate Speech Coding (HR) the required bandwidth is 8 kbits/s using the Least Significant Bit (LSB) of each PCM sample and for Full Rate Speech Coding (FR) and Enhanced

Full Rate Speech Coding (EFR) 16 kbits/s, using the two LSB of each PCM sample.

The speech quality is of special concern for a Mobile Switching Centre (MSC) based gateway. This is because of the number of encodings/decodings that can occur for IP based mobile calls.

For normal mobile calls we have the two following scenarios (using GSM as an example):

MS -> PSTN: GSM 06.10 - G.711
MS -> MS: GSM 06.10 - G.711 - GSM 06.10

When IP is a part of a call leg the following can happen when GSM 06.10 is used as the IP audio codec:

MS -> IP -> PSTN: GSM 06.10 - G.711 - GSM 06.1 - G.711
MS -> IP -> MS: GSM 06.10 - G.711 - GSM 06.1 - G.711 - GSM 06.10

Each encoding/decoding deteriorates the speech quality. For MS to MS calls we can hear the quality we get with two encodings. Adding one encoding as would be the case for an MS -> IP -> MS call will probably reduce the speech quality to an unacceptable level. One way to avoid this is by choosing G.711 as the IP_codec, but then no reduction in bandwidth is achieved, which should be one of the main goals with using IP.

Objects of the invention

A main object of the present invention is to improve the speech quality, especially for mobile VoIP calls, by reducing the encoding/decoding to a minimum for thereby avoiding deterioration of speech quality.

Another object of the present invention is to also reduce

the bandwidth, especially when using IP.

Another object of the present invention is to adapt the associated gateway (GW) so as to be in harmony with this optimisation.

Brief summary of the invention

The above objects are achieved in an arrangement as stated in the preamble, which according to the present invention is characterised by the features as stated in the enclosed patent claims.

In other words, the present invention suggests that the arrangement comprises means for either putting the TRAU in a transparent mode or letting the TRAU be bypassed altogether.

Further features and advantages of the present invention will appear from not only the enclosed patent claims, but also from the following description taken in conjunction with the enclosed drawings.

Brief disclosure of the drawings

Fig. 1 is a schematical diagram illustrating the principle of Tandem Free Operation (TFO).

Fig. 2 is a diagram illustrating the signalling sequences for TFO over IP.

Detailed description of embodiments

Fig. 1 is a schematical drawing illustrating the principle of Tandem Free Operation, and this principle has already been discussed on previous pages.

In Fig. 2 there is schematically illustrating signalling

sequences for TFO over IP, these signalling sequences illustrating one of several embodiments wherein the general idea of the present invention has been implemented.

In the following there will be given a more detailed description of how this embodiment can be implemented.

TFO over IP

The basic idea of the present invention is that no decoding is done in the TRAU. The TRAU can either be put in transparent mode or it is bypassed all together.

The MS encodes the speech either in Half Rate Speech Coding (HR), Full Rate Speech Coding (FR) or Enhanced Full Rate Speech Coding (EFR) and the speech samples are transmitted directly on to the IP network where they are assembled into Realtime Transfer Protocol (RTP) /User Datagram Protocol (UDP) packets. For HR the LSB bit on PCM is used, and for FR and EFR the two LSB bits are used.

The VoIP GW has to perform some Error Concealment. This is anyhow a normal function of a VoIP gateway.

Preferably this can be handled by standard #7 signalling using standard parameters, possibly using spare fields.

On the DTAP/BSSMAP side the BC (Bearer Capability) field is used, which is read by the Mobile Switching Centre (MSC) and mapped transparent to the User Service Information (USI) field on ISUP towards the VoIP Gateway.
BC contains two fields for negotiation during call set-up. If the GW does not support the preferred BC, the default is applied.

The preferred field could contain "TFO wanted", and the default field "no TFO". In BC two spare bits in octet 3a could be applied.

The VoIP gateway must have the final decision on which codec to use since it must terminate the correct TRAU frames when TFO is active.

For an outgoing call using TFO we could have the following scenario:

1. The user dials prefix to the destination number to indicate that he wants the call to be routed over an IP network.

2. After b-number analyses the MSC modifies the BC in SETUP. The BC contains two BC fields, one is fallback and one is preferred. The preferred BC is encoded with "TFO wanted", and codec type.

3. The MSC transfers the modified BC fields to the USI field in the outgoing IAM towards the VOIP gateway. The gateway now decides TFO or not and reads the codec type. The answer is transferred back to the MSC in the ACM message.

4. The MSC uses ASSIGNMENT REQUEST to request the wanted radio resources from the BSC. Information requiring TFO could be coded in fields like Channel Type, Classmark or others.

5. If TFO can be supported by the BSC this is confirmed back to the MSC in ASSIGNMENT COMPLETE. The BSC puts the TRAU in either transparent mode or bypassed mode. The method chosen here is of local (BSC) relevance only.

6. The MSC continues the call setup with the relevant DTAP messages.

7. If TFO was not accepted by the VoIP gateway the call is handled like a "no TFO" call.

Note:

- In this scenario the codec type is decided by the gateway in case of TFO.
- Fallback from TFO during speech is not considered necessary in a basic implementation.

For incoming TFO calls a reverse scenario applies. In case TFO is not supported by the BSC, or the MS does not support the incoming IP codec, the VoIP gateway must support fallback to G.711 speech.

Abbreviations

ACM	Address Complete Message
BSC	Base Switching Centre
BC	Bearer Capability
BSS	Base Station Subsystem
D-AMPS	Digital-Advanced Mobile Phone Service
DSP	Digital Signalling Processor
DTAP	Direct Transfer Application Part
ETSI	The European Telecommunications Standards Institute
EFR	Enhanced Full rate speech coding
FR	Full Rate speech coding
GSM	Global System for Mobile communication
GW	Gateway
HR	Half Rate speech coding
IAM	Initial Address Message
IP	Internet Protocol
ISUP	ISDN User Part
LSB	Least Significant Bit
MAP	Mobile Application Part
MS	Mobile Station
MSC	Mobile Switching Centre
PCM	Pulse Code Modulation
PDC	Personal Digital Cellular
PSTN	Public Switched Telephone Network
RTP	Realtime Transfer Protocol
SPE	Speech Encoding Eq
SPD	Speech Decoding Eq
TFO	Tandem Free Operation
TRAU	Tranceiver and Rate Adapter Unit
UDP	User Datagram Protocol
USI	User Service Information
VoIP	Voice over IP
ITU	(International Telecommunications Union, Geneva, Switzerland) formerly the CCITT (Consultative Committee for International Telephony and Telegraphy) is an international organization founded

in 1865 and headquartered in Geneva that sets communications standards.

P a t e n t c l a i m s

1. Arrangement for improving the speech quality, especially for mobile VoIP (Voice over IP) calls, which arrangement comprises a Transceiver and Rate Adapter Unit (TRAU) in which an encoded speech signal from a Mobile Station (MS) is transcoded, *c h a r c t e r i z e d i n* that said arrangement comprises means for either putting the TRAU in a transparent mode or letting the TRAU be bypassed altogether.

2. Arrangement as claimed in claim 1, *c h a r c t e r i z e d i n* that the MS in question comprises means for encoding the speech either in Half Rate Speech Coding (HR), Full Rate Speech Coding (FR) or Enhanced Full Rate Speech Coding (EFR), and that the speech samples are transmitted directly on to the IP network where they are assembled in to Real Time Transfer Protocol (RTP)/User Datagram Protocol (UDP) Packets.

3. Arrangement as claimed in claim 1 or 2, *c h a r c t e r i z e d i n* that TFO (Tandem Free Operation) over IP is achieved by using DTAP, BSSMAP and ISUP between the MS and the VoIP Gateway.

4. Arrangement as claimed in claim 2 or 3, *c h a r c t e r i z e d i n* that for HR the LSE bit on the PCM is used, and for FR and EFR the two LSE bits are used.

5. Arrangement as claimed in any of the claims 1-4, *c h a r c t e r i z e d i n* that the VoIP GW (gateway) in question comprises means for effecting a possible error concealment, as known per se, said error concealment being implemented by standard #7 signalling using standard parameters, possibly using spare fields.

6. Arrangement as claimed in any of the preceding claims,

c h a r c t e r i z e d   i n that on the DTAP/BSSMAP side the BC (Bearer Capability) field is used, said field being read by the Mobile Switching Centre (MSC) and mapped transparent to the User Service Information (USI) field on ISUP towards the VoIP gateway.

7. Arrangement as claimed in any of the preceding claims, *c h a r c t e r i z e d   i n* that the BC in question is adapted to contain two fields of negotiation during call set-up, one field for example being a fallback and the second field being preferred, and that if the GW in question does not support the preferred BC, then a default is applied.

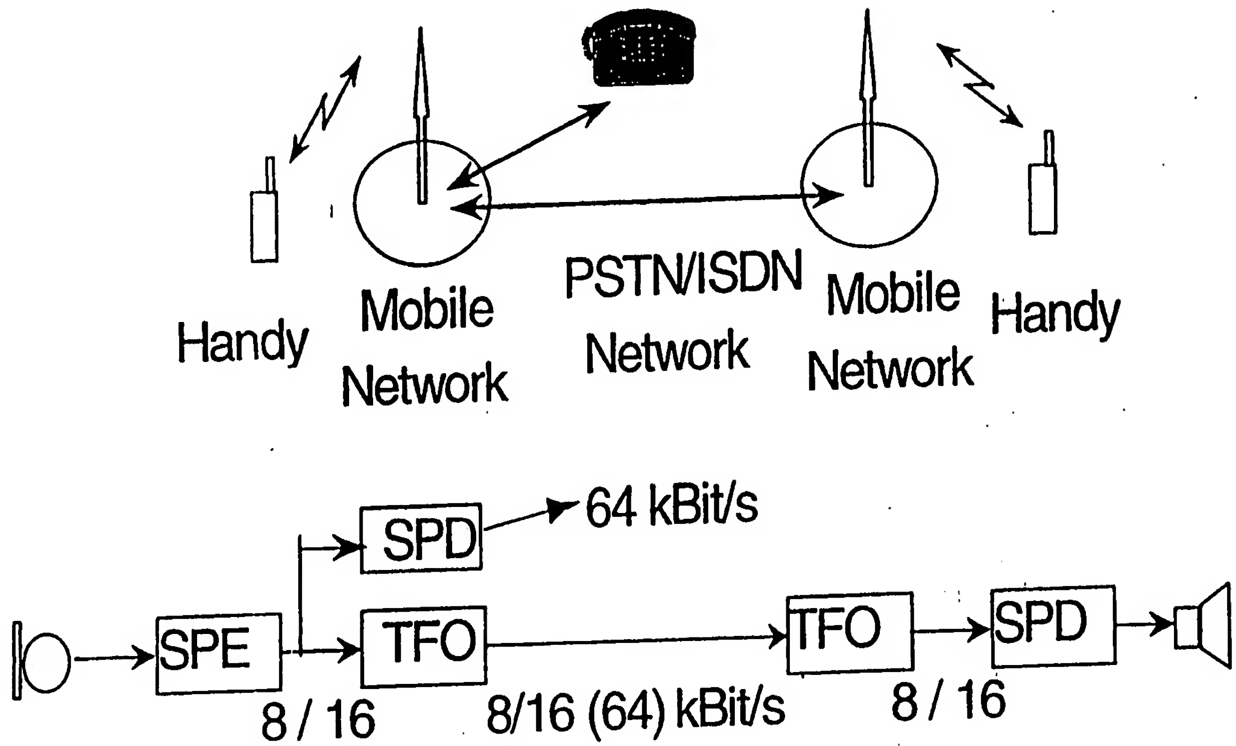
8. Arrangement as claimed in claim 7, *c h a r c t e r i z e d   i n* that the preferred BC field is adapted to contain "TFO wanted", and the default field "no TFO", and that in said BC there could be applied two spare bits in octet 3a.

9. Arrangement as claimed in any of the preceding claims, *c h a r c t e r i z e d   i n* that the VoIP gateway to which the network in question is connected, is provided with means for making the final decision on which codec to be used, said gateway further comprising means for terminating the correct TRAU frames when TFO is active.

10. Arrangement as claimed in claim 9, *c h a r c t e r i z e d   i n* that for an outgoing call using TFO the user may dial prefix to the destination number to indicate that the call should be routed over an IP network, that after b-number analyses the MSC modifies the BC in SETUP, for thereby encoding the preferred BC with "TFO wanted", as well as codec type, whereafter said MSC transfers the modified BC fields to the USI field in the outgoing Initial Address Message (IAM) towards the VoIP gateway, so that said gateway can decide TFO or not and read the codec type, the answer thereof being transferred

back to the MSC in the Address Complete Message (ACM) message.

11. Arrangement as claimed in any of the preceding claims 1-9,
c h a r c t e r i z e d   i n that for incoming TFO calls, and in case TFO is not supported by the BSC, or the MS does not support the incoming IP codec, then the associated VoIP gateway is adapted to support fallback to G.711 speech.



SPE: Speech Encoding Eq SPD: Speech Decoding Eq

Figure 1 Principle of Tandem Free Operation.

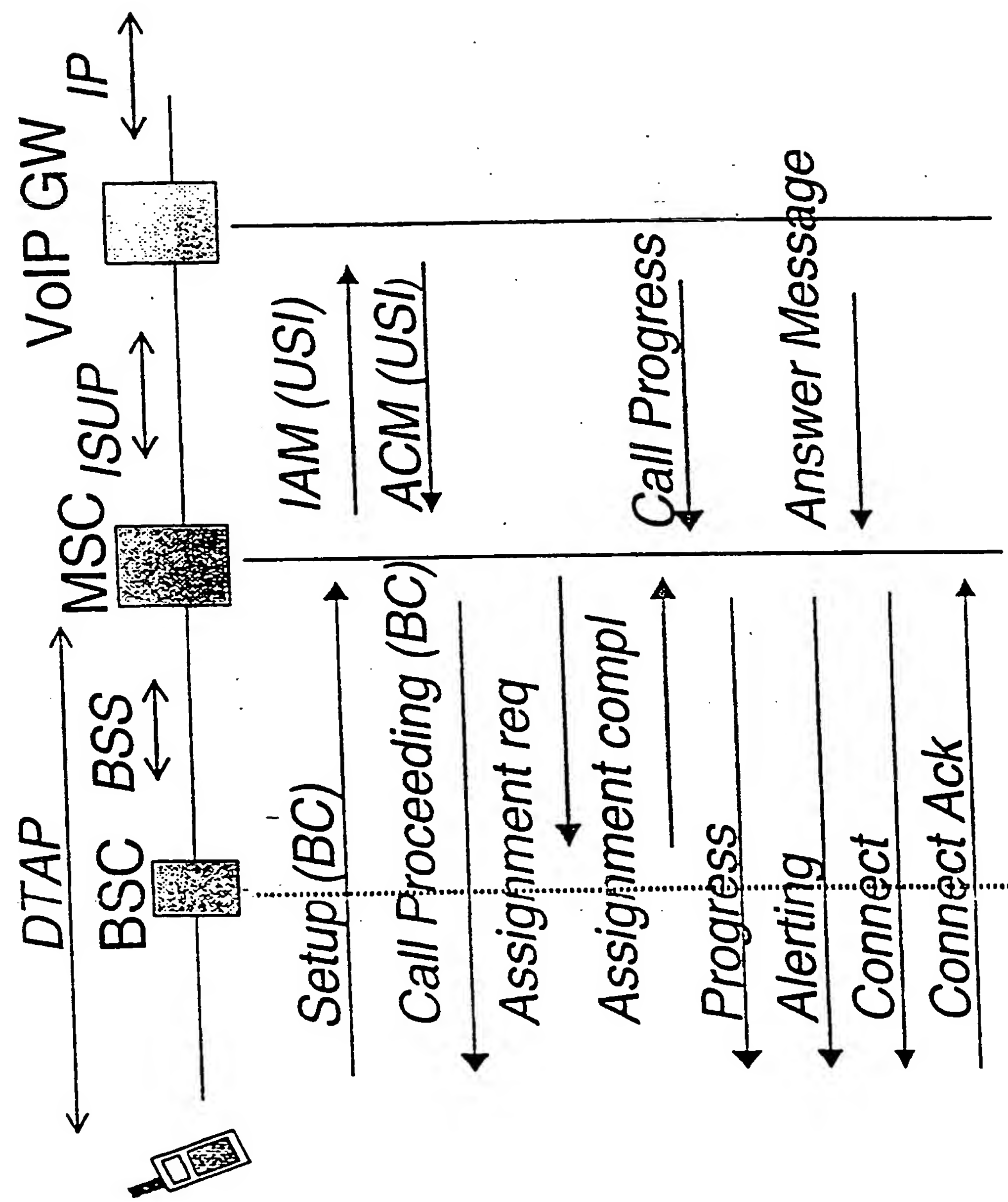


Figure 2 Signalling sequences for TFO over IP.

INTERNATIONAL SEARCH REPORT

International application No.
PCT/SE 00/00376

A. CLASSIFICATION OF SUBJECT MATTER

IPC7: H04Q 7/30
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC7: H04Q

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

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Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5608779 A (LEV ET AL.), 4 March 1997 (04.03.97), column 1, line 56 - column 2, line 16 --	1
X	WO 9616521 A1 (NOKIA TELECOMMUNICATIONS OY), 30 May 1996 (30.05.96), page 3, line 25 - line 34 --	1
X	ETSI TS 101732 (ETSI), July 1999 (07.99) (Release 1998), chapter 4, chapter 6.3.3 --	1
E,X	WO 0033590 A1 (TELEFONAKTIEBOLAGET LM ERICSSON (PUBL)), 8 June 2000 (08.06.00), page 3, line 1 - page 4, line 5 -- --------	1-11

☐ Further documents are listed in the continuation of Box C. ☒ See patent family annex.

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Information on patent family members

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